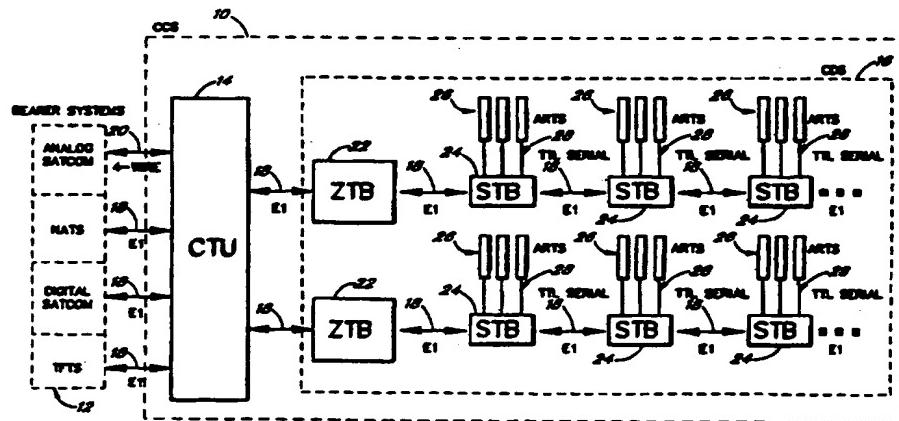


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(54) Title: MULTI-NODAL DIGITAL TELEPHONE DISTRIBUTION SYSTEM



(57) Abstract

A Cabin Distribution System (CDS) provides the necessary interfaces between each of the telephone units used by the passengers on the airplane and a Cabin Telephony Unit (CTU). The CTU is an intelligent telephony switch that controls and routes telephone calls between the passengers and a plurality of communication networks. A primary rate interface such as a CEPT E1 interface is used to connect the plurality of telephone units to the CTU in a loop-based configuration. The E1 interface provides thirty-two communication channels. Preferably, a link access protocol over the D-Channel is utilized to control the communications on the E1 interface. Utilizing the LAP-D protocol, of the thirty-two channels available on the E1 interface, a first channel is used for framing the communication messages, and a sixteenth channel, referred to as the D-Channel, is utilized as the data communication channel. Advantageously, the remaining thirty channels, which are referred to as B-Channels, are available to connect telephone calls. Data information transferred on the D-Channel is interrupted by each telephony group for a delay of sixteen frames to determine if the data is intended for the group. Voice information that is transmitted along the B-Channels is only interrupted for one frame by each group to determine if the data is intended for the group. By preventing the groups of telephones from interrupting the transmissions on the B-Channel for an extended length of time, unnecessary and unwanted delays in the voice communication are eliminated.

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MULTI-NODAL DIGITAL TELEPHONE DISTRIBUTION SYSTEMBackground of the Invention

The present invention relates generally to the field of digital telephone distribution systems.

5 It is desirable to implement a telephone distribution system which reduces the number and length of the wires required to connect a multitude of telephone units to a telephony switching device, such as a Private Automatic Branch Exchange (PABX). Traditionally, a "star" topology has been used with a PABX at the center with radial connections to each telephone unit. However, the "star" topology is not practical for
10 all situations, because of the number and length of wires that are required.

Further, in order to enable a multiplicity of telephone units to communicate with a central telephony switching device, and vice versa, a communication protocol which enables point to multi-point communication must be selected. Common networking protocols of the prior art, such as Ethernet and Token Ring protocols, enable point to
15 multi-point communication but are optimized for data transmission and not for voice delivery. These data transmission networks are undeterministic in their information delivery rates. The undeterministic quality of the transmission rates of these types of networks is not commonly a problem in data transmission systems, because it is not critical to the data receiver that the data be received at a specific instant in time.
20 However, these variations in delivery rate are detrimental to voice transmission, because they result in undesirable delays in the conversation.

Therefore, it would be desirable to provide a telephone distribution system which connects a number of telephone units to a central telephone control unit and does not suffer from unwanted delays in the transmission of the voice information through the
25 system.

Summary of the Invention

A multi-nodal digital telephone distribution system of the preferred embodiment allows multiple telephone or data communication devices to use a single connection to
30 a main digital telephony switching device, such as a Private Automatic Branch Exchange (PABX). A telephone communication system utilizing the preferred

embodiment of the multi-nodal digital telephone distribution system of the present invention comprises a main telephony switching device, and a plurality of telephony control units, wherein each of the telephony control units is connected to at least one telephone unit. A data link, comprising at least one data channel and a plurality of voice channels, connects the telephony control units to the main telephony switching system in a daisy chain connection. The multi-nodal telephone distribution system of the preferred embodiment is based on a "loop" topology, where at least one daisy chained connection of the telephony control units is connected to the telephony control switch, which is a node on the loop. In addition, a detecting circuit is connected to the data link to determine if the at least one data channel and at least one voice channel is to be received at the telephony control unit. The detecting circuit delays the transmission of at least one data channel without delaying the transmission of the at least one voice channel.

A method of controlling the multi-nodal digital telephone distribution system is disclosed. In a preferred embodiment, one data channel on the data link is used to connect each of the telephony control units to each other and to the main telephony switching device, thus creating the "loop" topology. Control data from the telephony control unit or main telephony switching device, is transmitted on the data channel of the data link to each of the telephony control units. The data channel at each of telephony control units is sampled to receive the information on the data channel. The sampled control data on the data channel is decoded to determine if the data is destined for the telephony control unit that intercepted the data. If the message is intended for the intercepting telephony control unit, the unit retains the message for processing. If the message is not intended for the intercepting telephony control unit, it sends it on to the next telephony control unit in the daisy chain. The control data from the last telephony control unit in the daisy chain is sent to the main telephony switching device on the data link for processing by the main telephony control switch. As each telephone unit checks the message to determine if the message is intended for the unit, the message is delayed. Delaying the data channel is not significant, as the user is usually unable to detect the delays in call set up time and cancellation. However, if similar delays were allowed on the voice channels, the user would detect such delays

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in the conversation. Therefore, once the data channel sets up a telephone call, it assigns one of the voice channels to the telephony control unit. The voice information received by the main telephony control switch is transmitted on the assigned voice channels of the data link to the telephony control unit that initiated the call without being intercepted by the other telephony control units. This direct connection of the voice channel with the telephone control units prevents the delays in the voice communication which exist in the data channel.

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In the preferred embodiment, the digital telephone distribution system is implemented in a commercial airplane for passenger use. The reason for the multi-nodal approach in a commercial airplane environment is to reduce the amount of wire required to connect a multitude of stations to a PABX. The reduction of wiring is important in an aircraft environment, where wiring space and weight allowance is limited. Further, the installation of a phone system into an airplane is a very complicated task due to the close quarters in an airplane. Therefore it is desirable to 15 reduce the amount of wires that are used to connect the telephone system.

Brief Description of the Drawings

Figure 1 a block diagram of a preferred embodiment of a telephone distribution system.

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Figure 2 is a block diagram of a Cabin Telecommunication Unit connected to a Zone Telephony Box which is in turn connected to a plurality of Seat Telephony Boxes.

Figure 3 is a diagram of the layers of the communications protocol on the D-channel.

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Figure 4 is a block diagram of the circuitry of the Seat Telephony Box.

Figure 5 is a block diagram of the circuitry of the Seat Telephony Module.

Figure 6 is a diagram of the software layers implemented in the Seat Telephony Module.

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Figure 7 is a ladder diagram which illustrates an example of the steps that occur from call initiation to call completion.

Detailed Description of the Preferred Embodiments

Figure 1 illustrates a block diagram of an exemplary digital communication system in which the multi-nodal digital telephone distribution system is implemented. In the preferred embodiment, the multi-nodal digital telephone distribution system is 5 implemented in a commercial airplane for passenger use. In the commercial aircraft environment, the overall digital communication system is referred to as a cabin communication system (CCS) 10, which is designed to provide high quality voice and data communication to commercial aircraft passengers. The architecture of the CCS 10 preferably connects multiple voice and fax/data channels to a plurality of 10 receivers/transmitters 12 on board the aircraft which establish a connection to worldwide communication networks, or bearer systems utilizing the preferred embodiment of the telephone distribution system. Preferably, the CCS 10 provides access to multiple bearer radio systems via the on board receivers/transmitters 12 such 15 as a North American Telephone System (NATS), Satellite Telecommunication Systems (SATCOM), and Terrestrial Flight Telephone Systems (TFTS). The CCS 10 preferably comprises a cabin telecommunication unit (CTU) 14 and a cabin distribution system (CDS) 16. The CDS 16 provides the necessary interfaces between each of the telephone units used by the passengers on the aircraft and the CTU 14. The CTU 14 is an intelligent telephony switch that controls and routes telephone calls between the 20 passengers and the bearer systems 12.

The CTU 14 is a digital telephony switching device which controls and routes calls to and from the telephone units on the aircraft to one of the external bearer systems. The CTU 14 incorporates Private Automatic Branch Exchange (PABX) functionality which is commonly used in multiple line conventional telephone 25 communications. The CTU 14 provides the necessary interfaces to communicate voice and data transmissions from the internal aircraft telephone system to the external bearer systems via a Primary Rate Interface (PRI). Preferably, the CTU includes approximately three to fifteen Counsel of European Posts and Telegraph (CEPT) E1 PRI interfaces 18 to communicate with the receivers/transmitters 12 of the digital bearer systems, such as the NATS, digital SATCOM, and TFTS systems, and at least one four-wire analog interface 20 to communicate with the analog transmitter/receiver of 30

the analog bearer systems, such as the analog SATCOM system.

As illustrated in Figure 1, the CDS preferably comprises at least one Zone Telephony Box (ZTB) 22 and may comprise up to eight ZTBs. Each ZTB 22 is connected to the CTU 14 via a data link 18. In a preferred embodiment, each ZTB 22 is connected to the CTU 14 via an E1 interface 18. Each ZTB 22 is connected to at least one telephony control unit, or Seat Telephony Box (STB) 24 via a data link, such as an E1 interface 18. In turn, each STB is connected to at least one Arm Rest Telephone (ART) 26 via a TTL serial connection and analog connections to the ear and mouth pieces on the lines 28. More preferably, up to three ARTs 26 can be connected to each STB 24. Advantageously, utilizing the CDS 16 of the present invention an ART 26 can be provided at every seat on the airplane. This enables a greater number of passengers to have access for initiating telephone calls than many prior art systems.

In the preferred embodiment, the ZTB 22, as illustrated in Figure 2, does not include any telephony electronics. The telephony electronics are located on the STBs 24 and on the CTU 14. The ZTB 22 is utilized for wire termination and power distribution to the STBs 24 and their respective ARTs (Figure 1). For wiring convenience the STBs 24 are connected in daisy chain loops to the ZTB 22. From the ZTB 22, a communication loop 30 is connected to a first STB 24 via an E1 interface 18; and in turn, the first STB 24 is connected to a second STB 24 which is in turn connected to a third STB 24 which is in turn connected to further STBs 24 until the end of the chain is reached. At the last STB 24 in a chain, a loopback plug 32 is provided to connect the outgoing transmit messages from the STB 24 to return wiring connections in the STB 24. For convenience, the return wiring is routed through each of the STBs 24 via the return wiring connections to the ZTB 30 and ultimately to the CTU 14. However, as will be understood by those of skill in the art instead of looping the wiring through each of the STBs 24 it is possible to connect the output of the last STB 24 in each chain directly to the input of the CTU 14. However, in a typical airplane environment it is preferable to reduce the amount of wiring required, therefore the loopback method is preferred over the direct connection of the last STB 24 in the chain to the CTU 14.

Preferably, the ZTB 22 comprises up to four communication loops 30 of STBs

24 which are indicated on Figure 2 as STB Loop 0, STB Loop 1, STB Loop 2 and STB Loop 3. In the preferred embodiment, twenty STBs 24 may be connected to STB Loops 0 and 2 and twenty STBs 24 may be connected to STB Loops 1 and 3; thus forty STBs 24 are preferably connected to each ZTB 22. However, up to sixty four STBs 5 24 may be connected to and addressed by each ZTB. As will be recognized by one of skill in the art, additional STBs 24 may be added to each ZTB 22 by increasing the number of address spaces reserved to address the ZTBs 22.

As illustrated in Figure 2, it is possible to think of the communications from the CTU 14 to each of the STBs 24 and back to the CTU 14 as a single large loop which is routed through the ZTB 22. In addition, the ZTB 22 transfers power from a power supply 33 to each of the STBs 34 and their respective ARTs (Figure 1) along a set of power distribution wires. With regard to power distribution, the ZTB 22 preferably routes 115 VAC at 400 Hz from a power supply through the ZTB 22 to the STBs 24. The STBs 24 in turn provide power to their respective ARTs. In an alternate 10 embodiment, power is provided directly to each of the STBs 24 from the airplane's power supply. With regard to the communication between the CTU 14 and the STBs 24, an E1 transmission (Tx) pair from the CTU 14 is preferably routed through the ZTB 22 along the dotted internal connection line to STB Loop 0. In the preferred embodiment, the power lines are routed on a cable 34 which also contains the 15 communication wiring. The STBs 24 in each of the STB loops 30 are configured in a daisy chain, whereby the first STB 24 in STB Loop 0 receives data on its receiver (Rx) pair (from the CTU's Tx pair 35;) and sends data to the next STB 24 in the loop 30 on its Tx pair (to the next STB's Rx pair). The last STB 24 in a loop has a 20 loopback plug 32 attached to it that routes the Tx pair coming from the last STB 24 back through all of the STBs 24 in the loop 30 on a return cable 36 through passive connections on each STB 24. Once the Tx pair in STB Loop 0 is routed back to the ZTB 22, it is connected to the first STB 24 in the next STB loop 30 via a cable 34. The E1 transmission signal finally emerges from the last STB 24 in the last STB loop 25 30, such as STB Loop 3. The last E1 transmission signal is routed back to the CTU 14 on the ZTB's Tx pair in a return cable 36 to the CTU's Rx pair 38. Therefore, in 30 order to enable proper connection of the STBs 24 to the CTU 14, if one of the STB

loops 30 on the ZTB 22 does not have any STBs 24 then a loopback plug 32 must be placed on the ZTB to STB loop connector to complete the E1 circuit.

As discussed above, a primary rate interface such as a CEPT E1 interface is used to connect the CTU 14 through the ZTBs 22 to the STBs 24. The CEPT E1 interface is based on utilizing two twisted pairs of wires to connect two devices to each other thus forming a point-to-point interface. The clocking, and bit stream protocol for the CEPT E1 interface is described in Recommendation G.703 of the International Telegraph & Telephone Consultative Committee (CCITT), as is well known to those of skill in the art and is hereby incorporated by reference. The interface is run at 2048 Kilo-bits per second (Kbps). The bit stream is broken up into 32 channels, each providing 64 Kbps throughput. Of the thirty-two channels available on the E1 interface, a first channel is used for framing the communications, i.e., delimiting each frame. The sixteenth channel on the E1 interface, also referred to as the D-channel, is utilized as the signaling link between the CTU and each of the STBs to request and negotiate call setup or call clearing and other signaling activities. The other thirty channels, referred to as Bearer channels or B-channels, are used as communication links to transfer voice, voice-band data, and packet data from the ARTs to the CTU and ultimately to the bearer systems. Advantageously, by utilizing the E1 interface up to thirty of the ARTs may be utilized at the same time, because thirty B-channels are available to connect telephone calls. This is a significant advantage over the prior art systems which enabled fewer telephone units which shared the same wiring to be operated at the same time.

In order to keep track of the communications on the D-channel, link access communications protocol over the D-Channel (LAP-D) is utilized. As illustrated in Figure 3, the traditional model for depiction of a communication protocol is to use a "stack" model, where each logical layer of the protocol is shown as a "box", with the stacked on top of each other, from the bottom, up. The communications protocol on the D-channel is broken up into a plurality of layers, the first layer (L1) or physical layer 44, the second layer (L2) or the data link layer 44 and the third layer (L3) or the network layer 46. As indicated above, the physical layer 44 is described by the CEPT E1 standard, which is hereby incorporated by reference. In the preferred embodiment,

the data link layer 45 of the protocol is described in the CCITT Recommendation Q.921, which is hereby incorporated by reference. The network layer of the preferred embodiment is described in CCITT Recommendation Q.931, which is hereby incorporated by reference. The above protocol standards which have been incorporated by reference are well known to those of skill in the art. The collection of the CEPT E1 physical layer 44, the Q.921 data link layer 45 and the Q.931 network layer 46 is sometimes referred to as the ISDN protocol. In addition to the standard portions of the protocol, the STBs 24 include a software layer which enables several STBs 24 on a loop to use the same physical medium to communicate with each other. This software program layer, which will be described in more detail later, is called a Medium Access Control (MAC) layer and provides the point to multi-point arrangement of the telephone distribution system. As described below in association with Figure 6, the MAC layer is located between the physical layer and the data link layer. The MAC layer performs the arbitration among all of the STBs 24 on the loop which want to use the D-channel for signalling.

Figure 4 illustrates a block diagram of the electronics resident on each of the STBs 24. The electronics provided on each STB 24 preferably comprises a Seat Power Module (SPM) 40 and a Seat Telephony Module (STM) 42. As described above, power lines 48 and E1 transmissions lines 18 are routed from the ZTB (Figure 2) to a first STB in a loop. The power lines 48 from the ZTB are routed to the SPM 40 which converts the A/C power received from the ZTB to DC power. The SPM 40 routes the DC power to the respective ARTs along line 50 and to the STM 42 along line 52. The E1 transmission lines 18 from the ZTB are routed to the STM 42. The STM 42 controls the telephony functions of the ARTs. Links are set up between each of the ARTs to the STM 42 to transmit and receive voice information, data and command information to/from the ARTs along lines 62 and 64. Outgoing E1 data from the STM 42 on lines 68 as well as the power from the Seat Power Module 40 on lines 48 are routed to an external connector on the STB 24 and are transmitted along lines 72, 48 respectively to the next STB in the chain or to the CTU if the STB is the last module in the chain.

Figure 5 illustrates a more detailed functional block diagram of the hardware on

the Seat Telephony Module (STM) 42. The E1 interface 18, i.e., the Rx transmission pairs 74 and the Tx transmission pairs 76, is connected to the STM 42 via a relay 78. The relay 78, when set, enables the E1 interface 18 to connect to the hardware of the STM 42. However, if the STM 42 is not functioning properly or if it is desired to disable the STM 42 functions for a group of users, the relay 78 can be opened and the E1 18 connection is terminated.

Assuming that the E1 interface 18 is connected through the relay 78, the information on the Rx pair 74 is received by E1 interface hardware 80. The E1 interface hardware 80 preferably comprises a framer 82 and an LIU 84. The input Rx signal on the line 74 is transmitted first to the framer 82. The framer 82, typically a crystal and associated logic, is utilized to differentiate each of the channels as they are received from the E1 interface 18. The Rx signal is then transmitted to the LIU 84, which converts the E1 signals from the voltage level required for E1 transmission to a voltage level that is acceptable to the hardware in the STM 42. After the E1 interface hardware 80 converts the bit stream received from the E1 interface 18 to signals that are understandable by the STM 42, the E1 interface hardware 80 sends the signals received on the D-channel to the CEPT E1 transmit and receive (XCVR) hardware 85. The CEPT XCVR 85 utilizes the High Level Data-link Control (HDLC) protocol, which describes how to assemble a collection of octets, i.e., 8-bit values, to form a complete HDLC message. The HDLC interrupt hardware acts in association with the XCVR 85 to interrupt a microprocessor 86 when a complete message is received. Preferably, the HDLC interrupt hardware is resident on the microprocessor 86. In the preferred embodiment, a Motorola 68302 microprocessor which has the HDLC interrupt hardware on the microprocessor is utilized.

Referring also to Figure 6, the CEPT XCVR 85 preferably comprises a receive (Rx) queue 87 and a transmit (Tx) queue 88. The Rx queue 87 collects the data received on the D-channel and groups the data into the HDLC message packet format which is readable by a microprocessor 86. The Tx queue 88 receives messages from the microprocessor 86 via the XCVR Tx queue service request routine (XCVR SQR) 89 and transfers the next available message in the Tx queue 88 into the appropriate bit stream format required by the transmission standard. Once a complete message has

been received by the Rx queue 87 on the CEPT XCVR 85, an XCVR initiated HDLC interrupt service routine (XCVR ISR) 90 is initiated. The MAC layer software 91 running on the microprocessor 86 determines if the message received from the D-channel is intended for this STM 42. If the message received by the STM 42 is intended for the STM 42, the XCVR ISR 90 stores the message until the microprocessor 86 is available to act on the message. If the message received is not intended for this STM 42, the XCVR ISR 90 on the microprocessor 86 routes the message to the XCVR SQR 91, for delivery to the Tx queue 88. The message is stored in the Tx queue 88 until the D-channel becomes available. Once the D-channel becomes available, the message is sent through the LIU 84 and framer 82 in order to adjust the voltage and clocking of the signal to the E1 standard of transmission, and the message is sent along Tx pairs 76 of the E1 18 to the next STB.

The MAC layer 91 may be implemented in at least two distinct operational modes: the store-and-forward mode and the polling mode. In the preferred embodiment, the MAC layer 90 is implemented in the store-and-forward operational mode. In the store-and-forward mode, each STB 24 intercepts all D-channel messages. When a complete HDLC message is received by the XCVR RX queue 87, the XCVR interrupt service routine 90 is invoked to move the message from the XCVR RX queue 87 to the MAC layer queue 91 in the microprocessor. The intercepted message from the upstream STB 24 (closest to the CTU) are looked at by the MAC layer 91. If the packet is a broadcast message, i.e., if the message is sent to all of the STBs 24, the message is sent to the L2 layer 92, and a copy of the message is put back on the XCVR TX queue 88 via the XCVR SQR 89 for transmission the downstream STB 24 (to the next STB). If the message is addressed to this STB 24, the MAC layer 91 sends it directly to the L2 layer 92. If the message is not addressed to this STB 24, it is put back into the TX queue 88 via the XCVR SQR 89. In all cases, the XCVR TX queue 88 will remove a message from the head of the queue (least recent) and send it out on the D-channel once it is available. Therefore, in the store-and-forward mode the messages on the D-channel will be delayed by each STB 24 for some period of time. Presently, the approximate delay caused by the STB 24 checking to see if the D-channel message is intended for it and then passing the information onto the next STB 24 is 16 frames.

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In an alternate embodiment, the MAC layer 91 is implemented in the polling mode. In polling mode, the XCVR hardware 85 will "spy" on the D-channel by assembling local copies of the messages received from an upstream STB 24, and passing the D-channel bit stream through unhindered. With each received message, an interrupt will invoke the XCVR interrupt service routine 90 which will send the message to the MAC layer 91. The MAC layer 91 will look for a polling message addressed to the STB 24. Any other message not addressed to this STB 24 is thrown away. Once a polling message addressed to this STB 24 is received from the CTU 14, the MAC layer 91 will instruct the XCVR hardware 85 to intercept all D-channel messages until further notice. It will also instruct the XCVR SQR 89 for the XCVR TX queue 88 to commence its transmission operation for the next queued message.

While in this mode the MAC layer 91 will retransmit all messages from upstream STBs 24 which are not addressed to this STB 24 by putting them on the XCVR TX queue 88. If a broadcast message is received, it is copied and passed to L2 layer 92 as well as forwarded to the next STB 24. If the broadcast message received is an order to go into store-and-forward mode by the CTU 14, then the MAC layer 91 will assume the Store and Forward mode until further notice. Otherwise, if a Polling message addressed to another STB 24 is received, the MAC layer 91 instructs the XCVR hardware 85 to go into "spy" mode, and instructs the XCVR SQR 89 for the XCVR TX queue 88 to cease transmitting any more messages. In the poling mode, the CTU 14 is constantly switching between each STB 24 as it polls for data. During the time period that the STB 24 is polled, the STB 24 is constantly transmitting its data until the CTU 14 polls the next STB 24.

Regardless of the ultimate operational mode of the MAC layer 91, during the system initialization, the CTU 14 utilizes the store-and-forward mode to broadcast a registration request to all the STBs 24. During the registration request, each STB 24 informs the CTU 14 that it is operable and where it is located on the loop. This is accomplished by each STBs 24 in turn sending their response to the registration request, along with the responses from the other upstream STBs 24 to the CTU 14. Once the registration is completed, all STBs 24 are assigned Terminating Endpoint Identifiers (TEIs), i.e., addresses on the loop. At this point, the MAC layer 91 will remain in the

store-and-forward mode for the remainder of the system operation, unless the CTU 14 indicates otherwise.

In order to enter the polling mode, the CTU 14 will broadcast a commence polling message to all STBs 24. The MAC layer 91 will then instruct the XCVR hardware 85 to only "spy" on the D-channel traffic, as described above. The MAC layer 91 will still assemble and send a copy of the received messages to the XCVR RX interrupt service routine 90, but it does this in parallel with other STB 24 which are also listening to the D-channel. Also in polling mode, the XCVR service request routine 89 for the TX queue 88 will be instructed not to send any other information downstream to other STBs 24.

Regardless of the mode that the MAC layer 91 is operating in, if the message is intended for the STB 24, the STB 24 transfers the message to the link Q.921 L2 layer 92. Once the L2 layer 92 is available, the L2 layer 92 acts on the message. If the message received by the L2 layer 92 also requires action by the Q.931 L3 layer 93, the message is stored until the microprocessor 86 has the opportunity to act on the Q.931 L3 layer 93 of the software. Once the L3 layer 93 becomes available, the L3 layer 93 acts on the message. If the message received by the Q.931 L3 layer 93 also requires action by the applications layer (not shown), then the microprocessor 86 saves the message until the applications layer is able to act on the message. Once the upper-most layer has completed the required action on the message, if a response to the CTU 14 is required, the response message is sent back down through the software layers until the MAC layer 91 is reached. The message is received at the MAC layer 91 and sent to the XCVR SQR 89 for transmission via the XCVR TX queue 88 to the E1 interface 18.

Referring back to Figure 5, in addition to the D-channel interfacing, the microprocessor 86 receives and transmits signals from the ARTs 26. Preferably, the microprocessor 86 is connected to each of the ARTs via a TTL UART 94. When a key on the telephony keypad is depressed on the handset, the ART sends a signal to the STB, which is received by the UART 94 on the STM 42. Once a signal is received by the UART 94, the UART 94 interrupts the microprocessor 86 with a message in an ASCII format from the ART. If the message received on the UART 94 from the ART

requires action by the CTU 14, such as a call initiation sequence or a call disconnect signal, the microprocessor 86 processes the UART message and sends an appropriate message to the CTU 14 on the D-channel, as indicated above. The additional hardware on the STM 42 is preferably used to transmit voice and modem data from the ART 26 through the E1 interface 18 to the CTU 14. Preferably, a coder/decoder (CODEC) 95 is connected to each of the ARTs at one end and to a Time Division Multiplexer (TDM) 85 at the other. Preferably, the CODEC 95 converts voice data, which is an audio signal, to a digital signal that can be processed by the TDM 98. The CODEC 95 also converts the digital signals received from the TDM 98 to audio voice signals, which are transmitted to the ARTs. Preferably, the TDM 98 controls the receipt and delivery of the voice data to/from the ARTs from/to the CTU 14. Further, a modem/fax port on each ART is connected to a CODEC loop detector 96 for translating the analog signals to digital signals and visa versa. The CODEC loop detector 96 is also connected to the TDM 98. Preferably, the TDM 98 is an Extended Phone Interface Chip (EPIC), such as a Siemens PEB-2055. Additionally, the microprocessor 86 is connected to external control logic 100, flash memory 101, and RAM memory banks 102. Finally, the STM 42 includes passive wiring that routes the return signals, i.e., receive (Rx) and transmit (Tx) pairs 104, which are routed from the last STB in the loop back to the ZTB. Therefore, the Rx pairs that are received by the STM 42 on the return route are immediately routed to the Tx pairs on the STM 42 and then to the next STB via the E1 interface. Ultimately, the message is routed back to the CTU.

When a call has been initiated, the CTU assigns one of the B-channels to the STM 42 for transmitting audio data to and from the ARTs. The CTU sends a message on the D-channel to the STB indicating the channel assignment. The TDM 98 on the STM 42 that initiated the call then activates a hardware connection between the CODEC 95 assigned to the ART that initiated the call and the E1 interface hardware 80 in preparation for the receipt of the communication from the assigned B-channel. The E1 interface hardware 80 on the STM 42 monitors the signals transmitted on the E1 interface 18 and looks for data on its assigned B-channel. Data bits on the B-channels that are not assigned to the STM 42 are transmitted through the STB with only a one frame delay which is caused by the E1 interface hardware 80 on the STM

42 checking each B-channel as it passes through the STB to determine if the channel is assigned to this STB. As soon as the information is received by the STM 42 on the assigned B-channel, it is immediately converted to the proper format, i.e., voltage and clock rate, by the E1 interface hardware 80 which can be received by the TDM 98.

5 The B-channel information is routed through the TDM 98 to the appropriate CODEC 95 that is connected to the ART that initiated the call. Preferably, the information on the B-channel is routed through the E1 interface hardware 80, the TDM 98, and the CODEC 95 hardware to the ART without significant delays. Voice data received at the ART is routed through the dedicated CODEC 95 to the TDM 98 and is transmitted from the TDM 98 to the E1 interface hardware 80 for immediate connection to the dedicated B-channel for receipt by the CTU. By proving direct routing of the incoming assigned B-channel signals through the hardware and by only intercepting the B-channel for one clock cycle per STB to determine if the information is intended for this STB, the voice information can be received and transmitted through the system without any notable delay to the user.

10 Referring also to Figure 7, a ladder diagram is provided to illustrate an example of the steps that occur from call initiation to call completion. Typically, the phone call is initiated on the ART by a passenger who completes a series of initiation steps, such as pressing the on button, entering a credit card number, etc. The initiation message 106 from the ART 26 is sent to the STB by signals received through the UART 94. The UART 94 interrupts the microprocessor 86, which receives the call initiation message 106 from the ART 26. The microprocessor 86 processes the information received from the ART 26 and transmits a phone call request message 108 to the D-channel Tx queue on the CEPT XCVR 85. Once the D-channel becomes available, the message from the CEPT XCVR 85 is transmitted through the LIU 84 and the framer 82 in order to adjust the voltage and clocking of the signal to the E1 standard of transmission. The phone call request 108 is then sent on the D-channel from the initiating STB 24 to the next consecutive STB 24 on the E1 transmit (Tx) pair. The next STB 24 reads in the message 108 on the D-channel through its receive (Rx) pair of lines. The message 108 on the D-channel is sent to the framer 82 and the LIU 84 to transfer the message to a voltage level and frequency that can be received by the

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STB 24. The message 108 is collected by the receiving queue on the CEPT XCVR 85. When a complete message has been received, the HDLC generates an interrupt to the microprocessor 86, which runs an interrupt service routine to decode the message 108. Once it is determined that the message 108 is not destined for this STB 24, the microprocessor 86 sends the message 108 to the transmit queue on the CEPT XCVR 85. Once the D-channel becomes available, the message 108 is sent through the LIU 84 and framer 82 and through the Tx pairs on the STB 24 to the next STB 24. This process of message checking by each STB 24 continues until the message is received by the CTU 14.

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The CTU 14 processes the initiate phone call request 108 and assigns one of the B-channels as the transmission media for the call. In this example, the CTU 14 assigns channel 17 to the call request of the initiating STB 24. The CTU 14 sends a message on the D-channel to the first STB 24. As described above, the first STB 24 checks to see if the message 110 on the D-channel is intended for its receipt. After the STB 24 determines that this message 110 is not intended for it, the first STB 24 sends the message 110 out on the D-channel to the next STB 24. The message 110 travels through each of the STBs 24 until it reaches the requesting STB 24.

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Once the message 110 reaches the initiating STB 24 on the D-channel, the microprocessor 86 on the STB 24 determines the message 110 is intended for it. The microprocessor 86 processes the message from the CTU 14 and determines that the B-channel on channel 17 is assigned to the call. The microprocessor 86 sends the channel assignment information to the TDM 85. The TDM 85 sends a message to the E1 interface hardware 80 indicating that all information received on channel 17 is to be directly routed to the TDM 85. The TDM 85 in effect sets up a hardware connection between the CODEC 95 assigned to the initiating ART 85 and the E1 interface hardware 80 in preparation for the receipt of the communication on channel 17.

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With this connection made, a message 112 is sent to the ART 26 to indicate a call can be made on the assigned channel. Next, the passenger dials the requested phone number 114 on the ART which is sent to the STB 24 via the TTL serial interface. The phone number 114 is received by the UART 94 and is sent to the

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microprocessor 86. The microprocessor 86 transfers the dialing information to the Q.931 software. The Q.931 software sends the dialing information to the Q.921 layer. The Q.921 layer sends the dialing information to the CEPT XCVR 85. From the XCVR 85 the data is sent to the E1 interface hardware 80 and then on the D-channel of the E1 18. As discussed above, the phone number message 116 from the initiating STB 24 is sent along the loop to next subsequent STB 24, where it is determined that the message 116 is not intended for that STB 24. The STB 24, in turn, sends the message 116 on to the other STBs 24 in the chain until it reaches the CTU 14. Once the CTU 14 receives the phone number message 116, it establishes a connection with the external bearer system. Once the connection is made, the CTU 14 transmits the connecting information 118 on the D-channel to the first STB 24, which passes the connecting information 118 on until the initiating STB 24 receives the connection information 118 on the D-channel.

The connection information 118 received by the initiating STB 24, informs the STB that the listener can use the assigned B-channel, i.e., channel 17. While the call is connected, all voice and data information 122 regarding the current call is sent to from the ART 26 and through each of the STBs 24 along the B-channel, i.e., channel 17, to the CTU 14 and vice versa until the call is disconnected. However, as discussed above, the STBs 24 that are not assigned to the B-channel immediately reroute the information on the B-channel to the next STB 24 and eventually to the CTU 14 causing an insignificant delay in the transmission of the voice information.

After the conversation has ended, the user initiates a terminate call sequence on the ART 26; for example, the user presses the END key on the handset. This termination message 124 is sent on the TTL serial channel to the STB 24. The data is received by the UART 94 on the STB which interrupts the microprocessor 86. The microprocessor 86 processes the message 124 from the ART 26. The microprocessor 86 transmits a terminate message 128 on the D-channel to the CTU 14. The message 128 is received by each of the subsequent STBs 24. Once the STB 24 determines that the message 128 is not intended for it, it will pass the message 126 along the chain until the message 126 is received by the CTU 14. The CTU 14 will terminate the call with the bearer system.

Next, the CTU 14 sends a message 128 to the STB 24 indicating that the reserved B-channel, i.e., channel 17 in this case, is disconnected and therefore available for other users. The first STB 24 receives this message 128 and determines that the message 128 is not intended for it. The STB 24 sends the message 128 to the next 5 STB 24 in the chain. Once the initiating STB 24 receives the message 128, the microprocessor 86 receives and process the message 128. The microprocessor 86 sends a signal to the TDM 85 and to the E1 interface hardware 80, indicating that the assignment of channel 17 to the STB 24 should be terminated. The TDM 85 will break the link between the CODEC 95 and the E1 interface 18 that was reserved for channel 10 17 transmissions.

Advantageously, the CDS of the present invention reserves one of the B-channels for each telephone conversation that is initiated. By preventing other STB's from interrupting the voice transmission on the B-channel for an extended length of time unnecessary and unwanted delays in the voice communication are eliminated. 15 Further, by using a primary rate interface, such as an E1 transmission interface, the data delivery rate is fixed, so the voice information is delivered at a predictable rate. By providing an interface with a known data delivery rate, problems associated with undeterministic delivery times of voice messages are eliminated. These undeterministic delays are common with other packet networks, such as Ethernet and Token Ring 20 systems. In addition, by eliminating the number of devices that interrupt the voice transmission, unwanted delays, which cause difficulty in telephony communication, are reduced to undetectable levels in the system. Other messages that are not a part of the audio conversation are delivered along a separate data channel, i.e., the D-channel, which is interruptable. However, such delays are generally acceptable in data delivery. 25 Further, the loop configuration of the Cabin Distribution System (CDS) reduces the number of wires required to connect the telephony system of the present invention.

The present invention may be embodied in other specific forms without departing from its spirit or essential characteristics. The described embodiments are to be considered in all respects only as illustrative and not restrictive. The scope of the 30 invention is, therefore, indicated by the appended claims rather than the foregoing description. All changes which come within the meaning and range of equivalency of

the claims are to be embraced within their scope.

WHAT IS CLAIMED IS:

1. A telephone communication system, comprising
 - a main telephone switching device;
 - a plurality of telephony control units;

5 a telephone unit connected to all of said telephony control units;
a data link comprising at least one data channel and a plurality of voice channels, connecting said telephony control units to said main telephony switching device; and
a detecting circuit connected to said data link which determines if said at least one data channel and said at least one voice channel is to be received at said telephony control unit, wherein said detecting circuit delays the transmission of said at least one data channel without delaying the transmission of said at least one voice channel.

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- 2. The telephone communication system as defined in Claim 1, wherein said telephony control units are connected to said main telephony switching device in a daisy chain connection.
- 15 3. The telephone communication system as defined in Claim 1, additionally comprising:
a plurality of transmitters/receivers connected to the main telephony switching device to establish a connection to a bearer system.
- 20 4. The telephone communication system as defined in Claim 1 wherein said data link is an E1 interface.
- 5. A method of operating a telephone unit, wherein said telephone unit is connected to a multi-channel communications cable comprising at least one data channel and a plurality of voice channels, said method of operating a telephone unit comprising the steps of:
delaying said at least one data channel to monitor the need to connect said telephone unit to one of said plurality of voice channels; and
transmitting said plurality of voice channels through said telephone unit with minimal delay during said delaying step.
- 25 30 6. A method of controlling telephone connections in an airplane telephone

communication system, wherein said communication system comprises a main telephone switching device, a plurality of telephony control units, each of said telephony control units connected to at least one telephone unit, a data link comprising at least one data channel and a plurality of voice channels connecting said telephony control units to said telephony switching device, said method of controlling telephone connections comprising the steps of:

5 transmitting control data from said main telephony switching device on said at least one data channel of said data link to the first telephony control unit;

10 intercepting said control data on said data channel at each of said telephony control units;

decoding said control data at each of said telephony control units to determine if said control data is destined for the telephony control unit that intercepted the control data;

15 sending said control data that is not intended for the intercepting telephony control units to a subsequent telephony control unit on said data channel;

sending control data from a last telephony control unit to said main telephony switching device on said data channel;

20 assigning one of said voice channels to one of said telephony control units for processing a telephone call;

transmitting voice information received by said telephony switching device to one of said telephony control units on said assigned voice channel; and

transmitting voice information received by one of said telephony control units to said telephony switching device on said assigned voice channel.

25 7. The method of controlling telephone connections in an airplane telephone communication system as defined in Claim 6, wherein said plurality of telephone units and said main telephony switching device are connected via said data link in a daisy chain.

30 8. The method of controlling telephone connections in an airplane telephone communication system as defined in Claim 6, further comprising the step of:

processing said control data that is destined for the intercepting telephony control

unit at said intercepting telephony control units.

9. The method of controlling telephone connections in an airplane telephone communication system as defined in Claim 8, further comprising the step of:

5 establishing a connection between said main telephony switching device and an external bearer system.

10. A multi-nodal digital telephone distribution system, wherein said telephone distribution system comprising:

a multi-channel communications cable comprising at least one data channel and a plurality of voice channels;

10 a plurality of telephone control units, wherein each of said telephone control units is connected via a transmitter/receiver device to said multi-channel communications cable; and

15 a medium access control software program in communication with said transmitter/receiver device on each of said telephone control units to arbitrate the access of the telephone control units to the data channel on said multi-channel communications cable.

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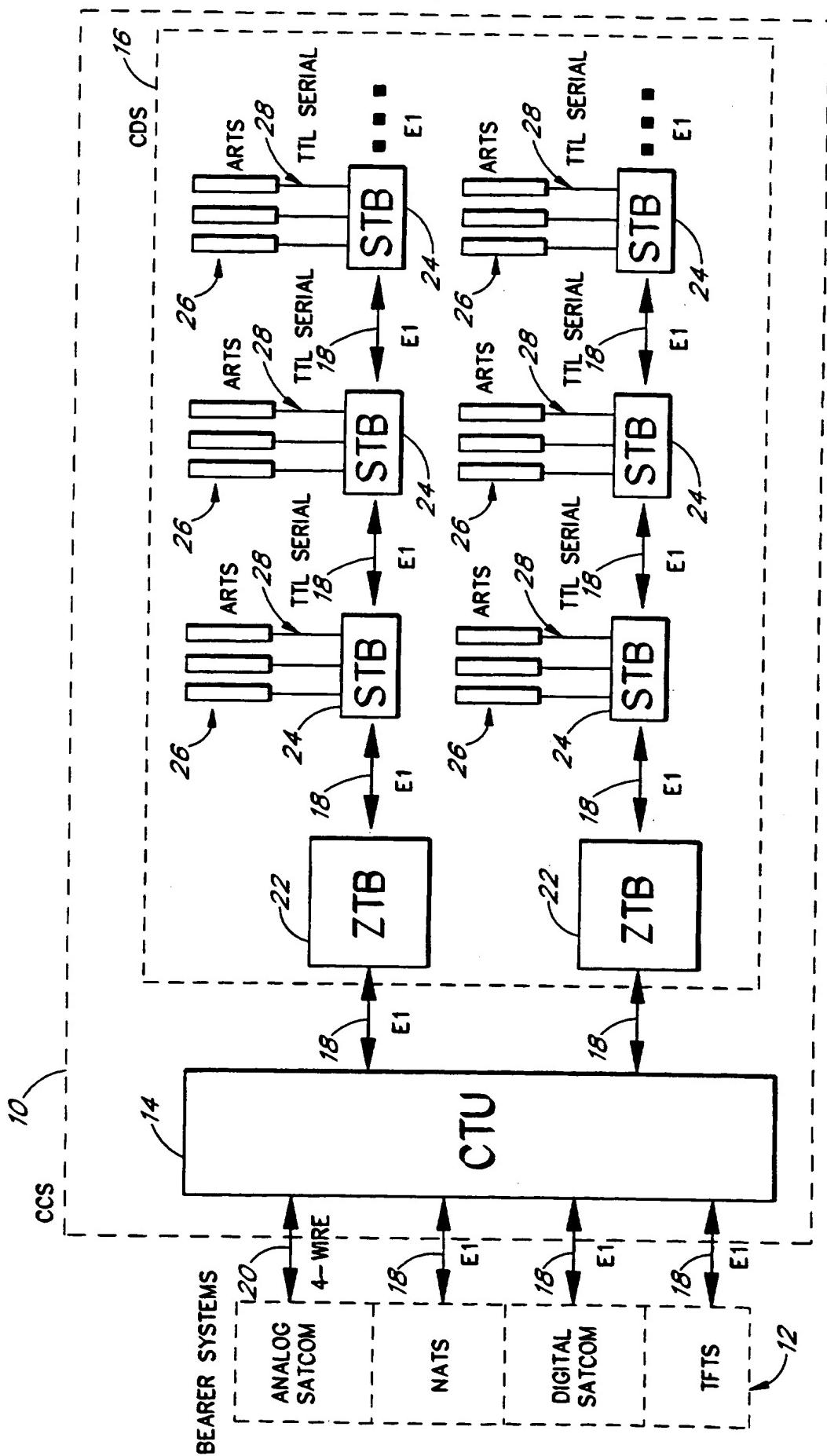


FIG. 1

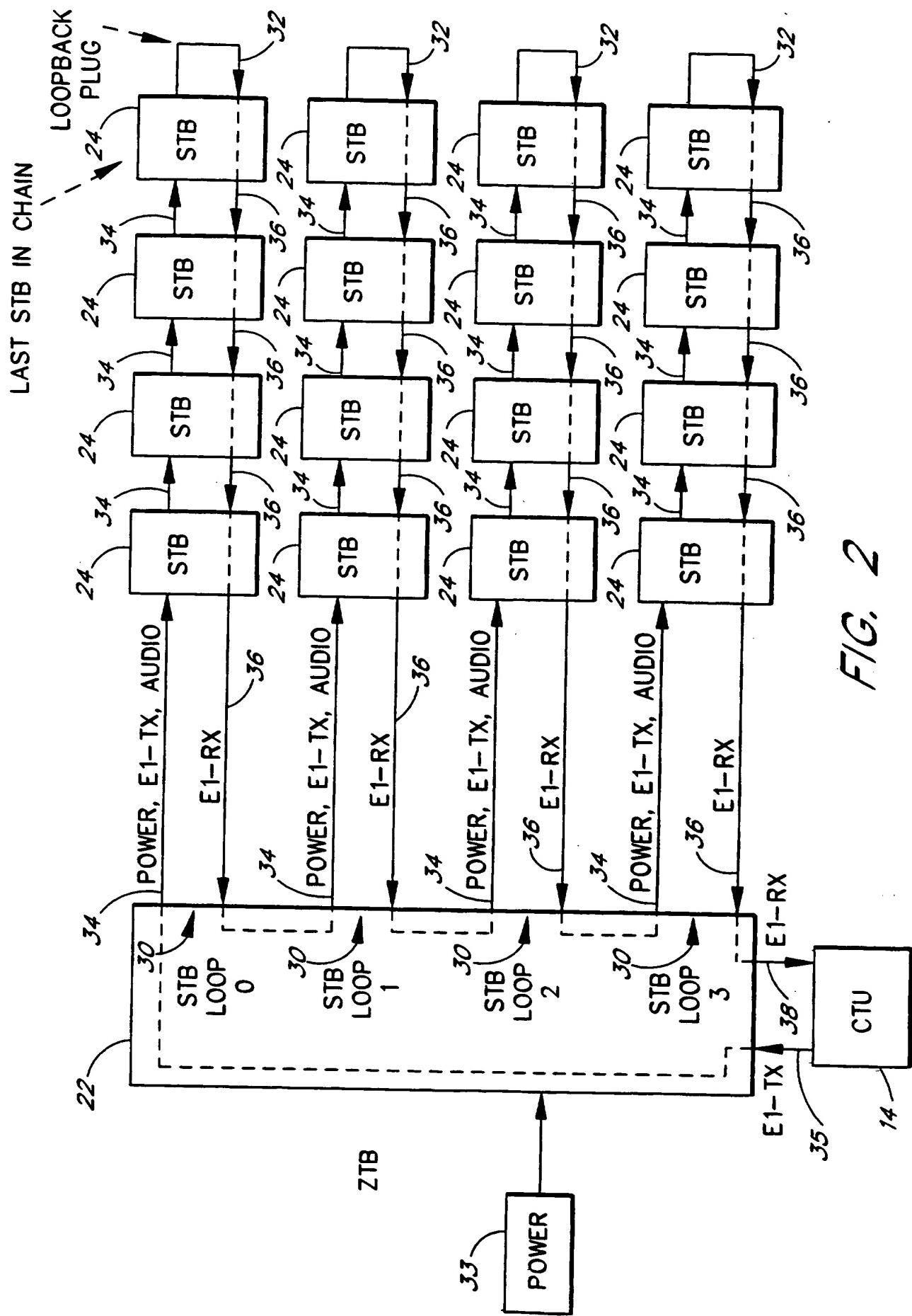


FIG. 2

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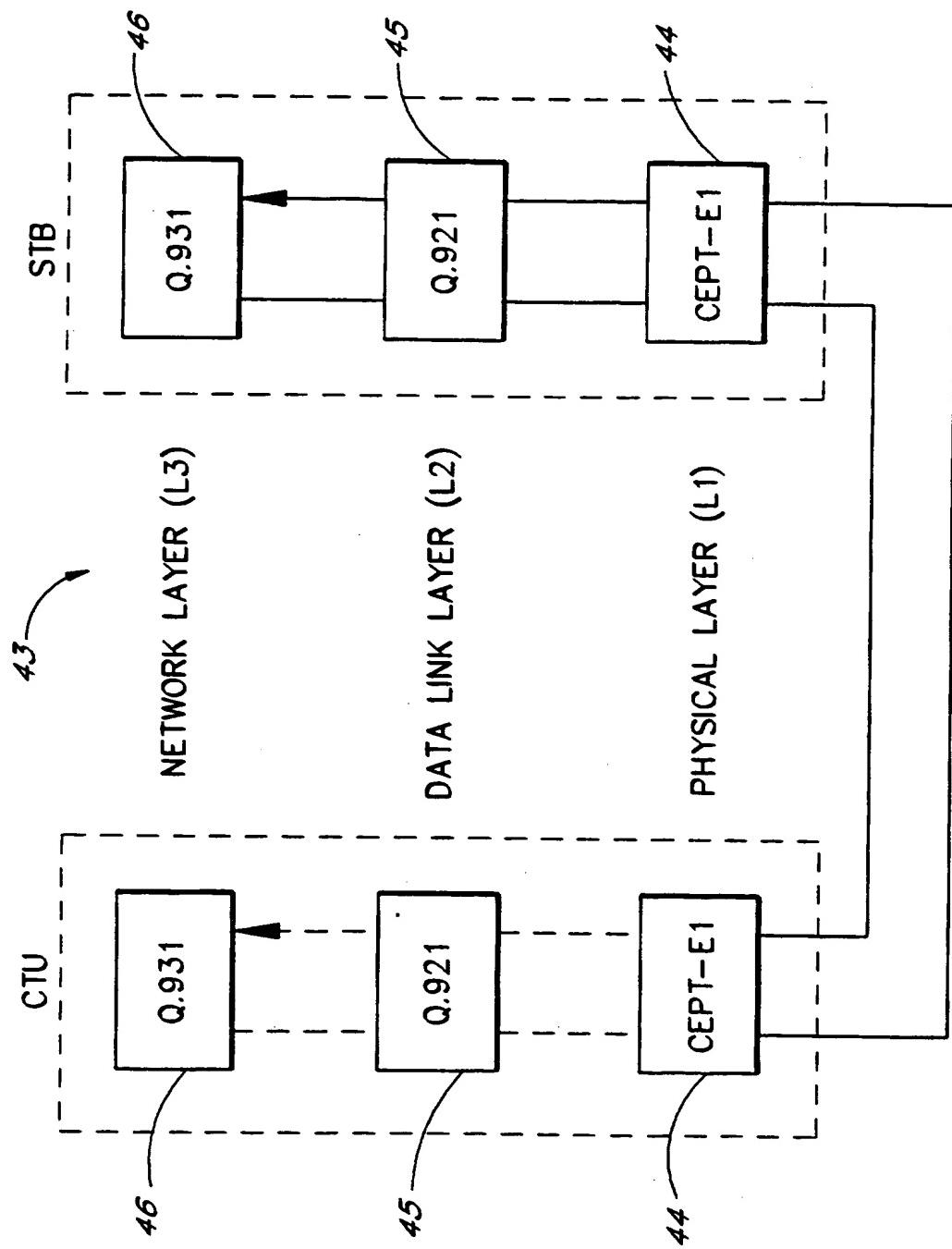


FIG. 3

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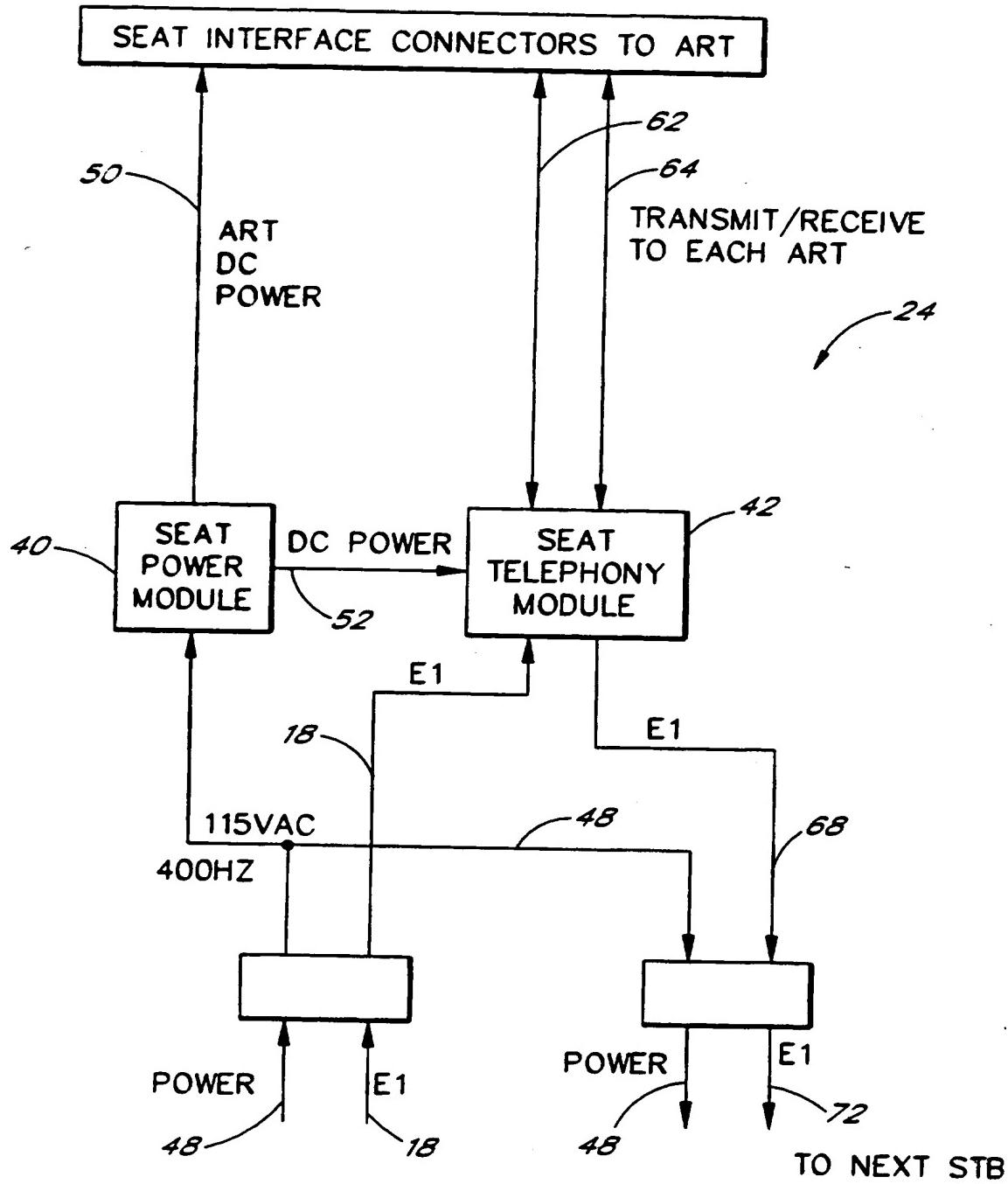


FIG. 4

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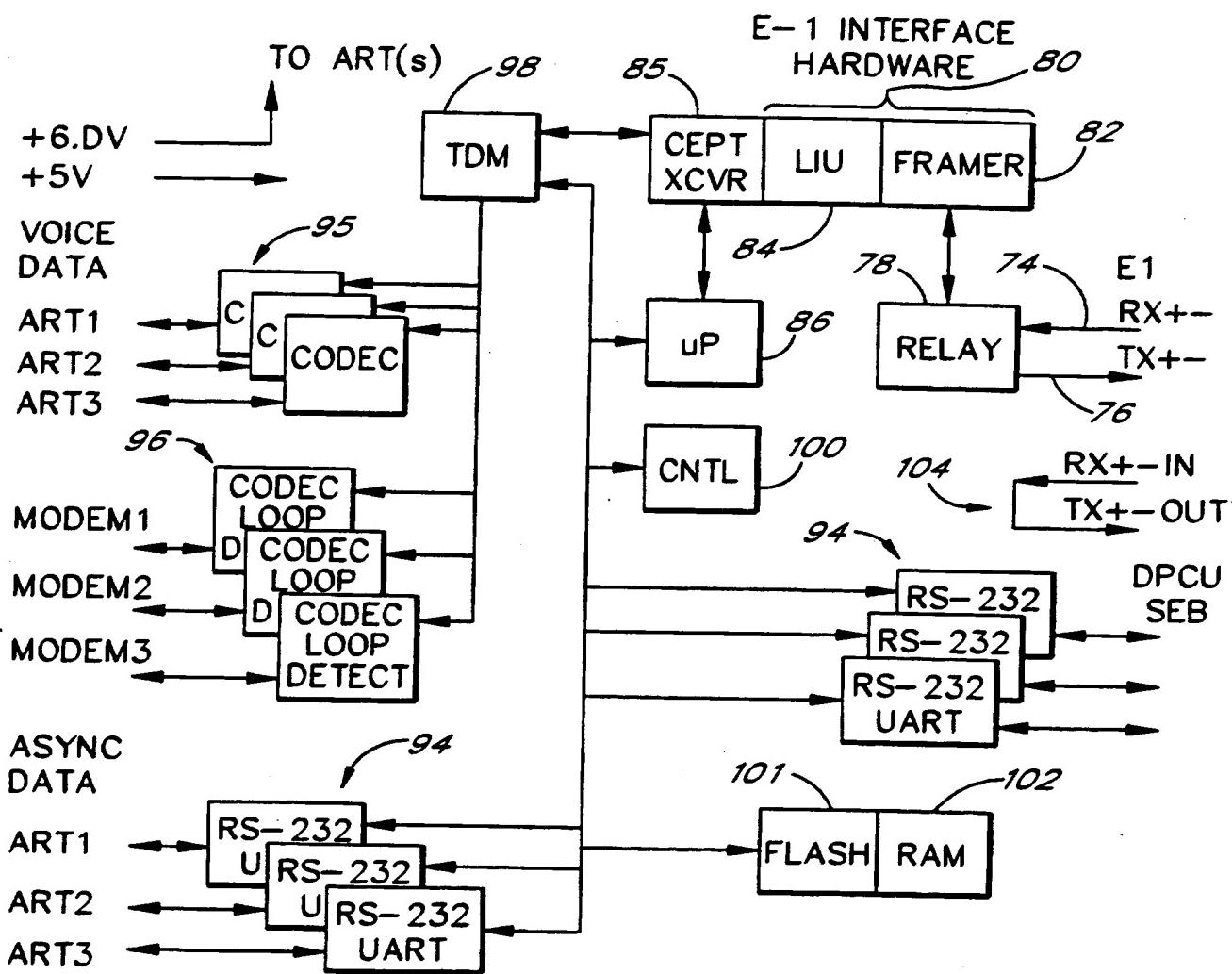


FIG. 5

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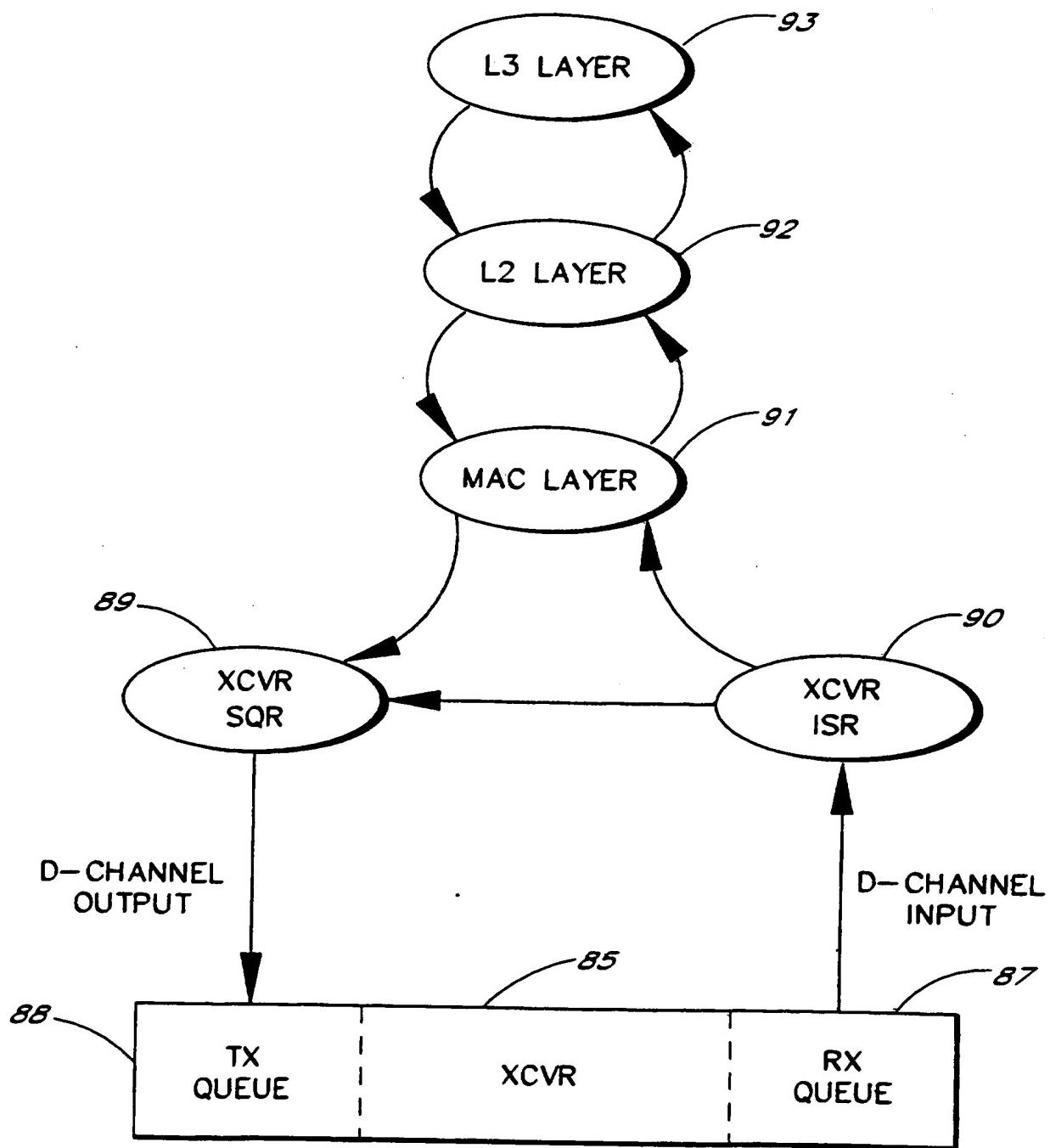


FIG. 6

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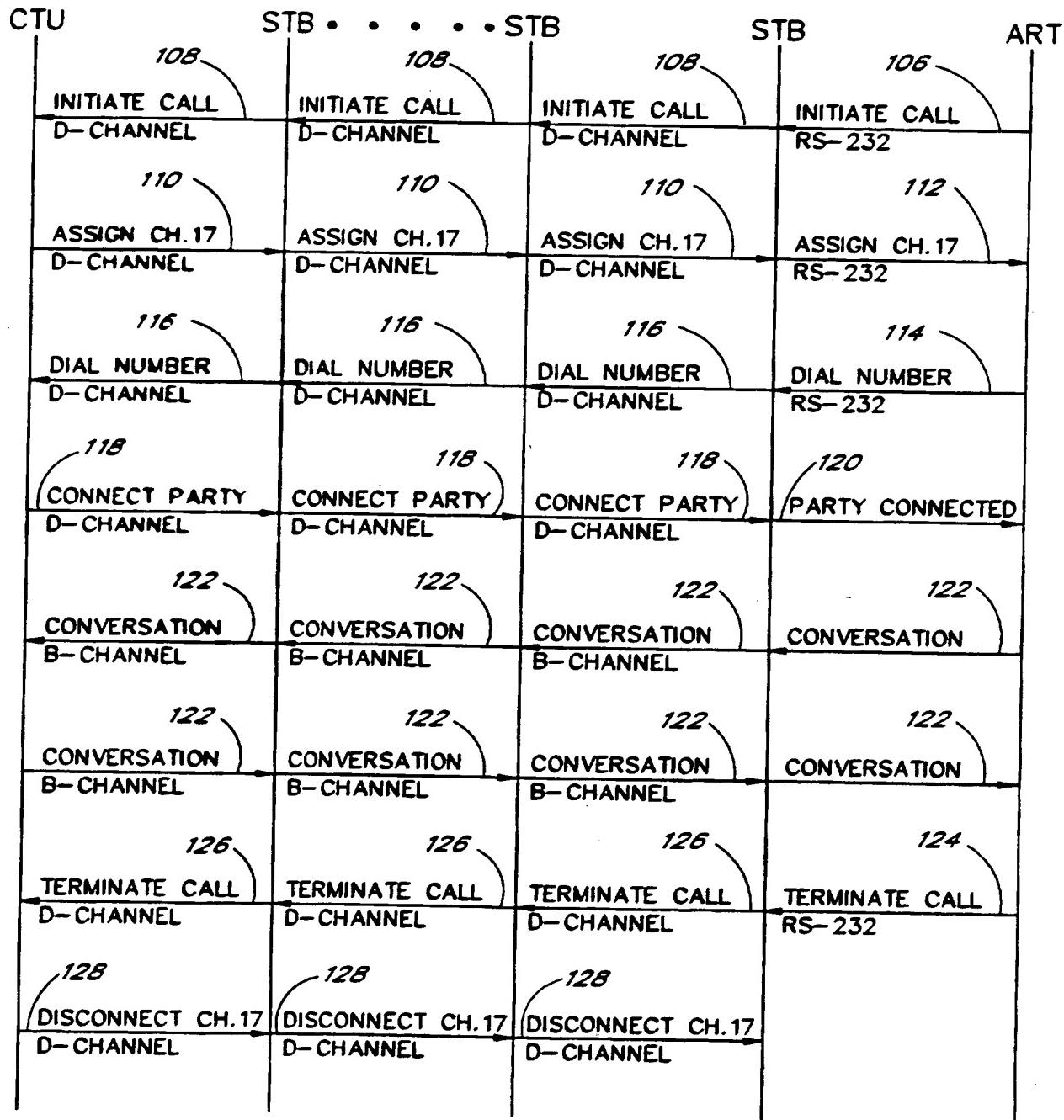


FIG. 7

INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 95/09218A. CLASSIFICATION OF SUBJECT MATTER
IPC 6 H04Q11/04 H04Q3/62 H04M9/02

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
IPC 6 H04Q H04M

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practical, search terms used)

C. DOCUMENTS CONSIDERED TO BE RELEVANT

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X	---	
A	AU,B,535 333 (L.M. ERICSSON) 15 March 1984 see page 6, line 26 - page 8, line 11; figures 2,3	10 1,2,5-7
X	---	
A	ELECTRICAL COMMUNICATION, vol. 60, no. 1, 1986 HARLOW, GB, pages 17-22, M.CORONARO ET AL. see page 19, left column; figure 2	10

	-/-	

 Further documents are listed in the continuation of box C. Patent family members are listed in annex.

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Date of the actual completion of the international search

31 October 1995

Date of mailing of the international search report

10.11.95

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INTERNATIONAL SEARCH REPORT

International Application No
PCT/US 95/09218

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